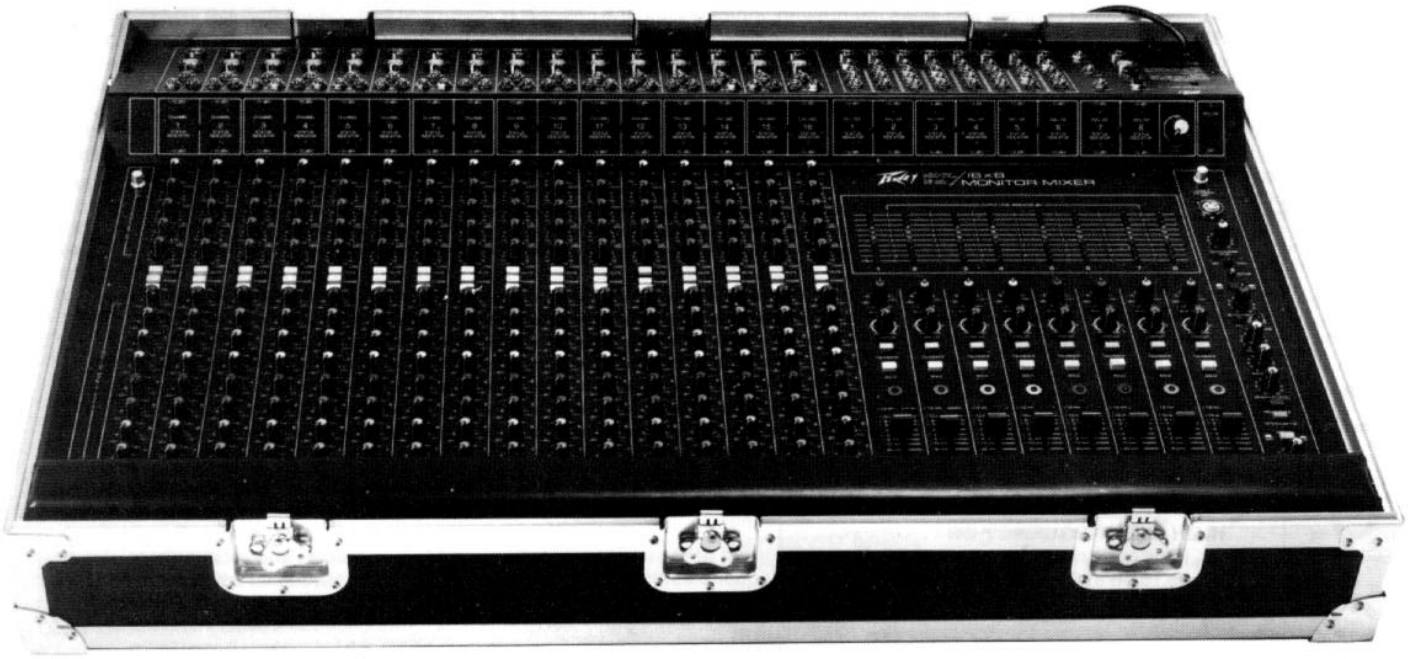


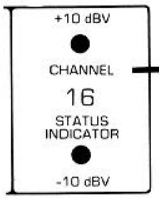
# Mark IV™ Monitor Mixer

## OPERATING GUIDE

Congratulations, you are the owner of one of the most functional, compact designs on the market today for monitor mixing. Many performers and groups continue to ask for several separate on-stage monitor mixes, but it seems that most players simply cannot afford the luxury.

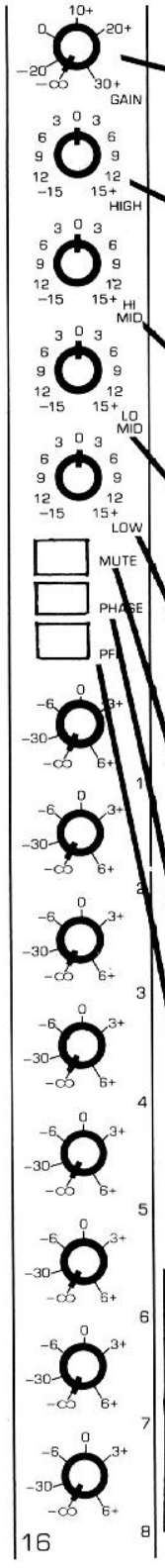
We have, with the Mark IV™, given you all the necessary tools for successfully accomplishing 8 totally separate monitor mixes and **second** made it "affordable." Please go over this manual thoroughly and be familiar with all the features before attempting the complex job of separate monitor mixing. Please be aware also that it takes "experience" to achieve a good monitor mix and don't expect to do a "perfect job" the first time you attempt to do a fully separate mix with the MK IV™ (or any other monitor mixer).

**WARNING**
**TO PREVENT ELECTRICAL SHOCK OR FIRE HAZARD, DO NOT EXPOSE THIS APPLIANCE TO RAIN OR MOISTURE.**
**BEFORE USING THIS APPLIANCE, READ BACK COVER FOR FURTHER WARNINGS.**




**LED STATUS DISPLAY**

The LED Status Display instantaneously displays the "operational state" of each respective channel. The -10 dBV c normal indicator is green and when illuminated, indicates that the channel is active and that a signal level of at least .3 volt RMS is being fed to the channel fader. The sampling of this status indication takes place immediately after the input gain control and also immediately after the equalization so that any boost or cut with the channel EQ will be exhibited by the LED indication. The +10 dBV (red) LED serves as a warning system and indicates that overload **could** occur. **The channel is capable of +18 dBV and just because the +10 LED flashes it doesn't mean that the channel has gone into clipping** (severe overload), but simply that +10 has been reached (3 volts). Absence of input signal for any reason will result in no illumination of the -10 dBV (green) LED. Proper adjustment of input gain should allow the +10 dBV LED indicator to only flash on extreme peaks of program material.




**INPUT GAIN CONTROL**

The input gain control of the Mark IV™ Monitor utilizes a dual control element configuration so that input attenuation and gain adjustment occur simultaneously. This arrangement allows the vital input circuitry to handle almost any input voltage from low level mics up to speaker levels. The status of this input preamp is monitored by the normal -10 dBV/+10 dBV (LED) indicators located above. The function and operational requirements of this input gain stage are conventional and should present no operational difficulty.

**HIGH FREQUENCY EQ**

This control determines the relative boost or cut of the high frequency adjustment on this console and features a 15 dB boost or cut at **15 KHz**. This control operates in a "shelving" configuration which prevents any additional gain above the useful audio spectrum from causing undesirable hiss and instability. The zero setting indicates a flat frequency response with no alteration of the extreme highs and clockwise rotation indicates a boost while counterclockwise indicates a cut or a dip in the high frequencies. Usually feedback is not much of a problem at very high frequencies but the control is very important for giving that "sibilant sound" on the extreme highs and on the S's and T's of vocals.

**HIGH MID FREQUENCY EQ**

This control determines the relative boost or cut of the upper mid or lower treble frequencies and is capable of plus or minus 15 dB at **3 KHz**. A balance of mid-range frequencies should begin with a flat setting of zero so that adjustments can be made with a cut or a boost of this control. The action of this control is peak/notch type which interfaces with the adjacent equalizers in a smooth manner to yield optimum combining capabilities to the channel equalization. Usually a minus setting of this control is desirable to eliminate harshness with most systems and to help control feedback.

**LOW MID FREQUENCY EQ**

This control is operational in the lower mid range or upper bass region and is capable of 15 dB boost or cut at **400 Hz**. This EQ control exhibits a peak/notch characteristic which operates smoothly with the adjacent equalizer controls and allows proper balancing of the upper bass and lower mid-range frequencies. This control is very vital for controlling the mid-range frequencies of the human voice when vocals are being amplified. Many sound reinforcement situations will require a slight negative setting of this control in order to eliminate muddiness in the vocal range and present a drier, more crisp **monitor** tonality.

**LOW FREQUENCY EQ**

This control is capable of 15 dB boost or cut at 60 Hz and demonstrates a shelving characteristic so that excessive lows are not boosted below the usable range of sound reinforcement speaker systems. Shelving action of this circuit has proven to yield much more satisfying and effective equalization characteristics than some of the "wide open" EQ circuits claiming 20 to 25 dB boost and cut. Care should be taken with the low frequency control to not overboost lows on any particular instrument or vocal due to the fact that tremendous amounts of power amp **headroom** will be used up and a general "muddiness" will be apparent in the monitor system. Woofers may also be blown much more easily with extreme low frequency boost.

**MUTE CONTROL**

The mute control is a push/push switch which enables the operator to switch off an entire channel and all of its functions simply be depressing this switch. For normal operation the mute switch should remain in the up (normal) position. Many times in sound reinforcement monitor applications it is necessary to **kill** or mute live microphones which are on stage but are not being used in any of the monitor mixes for a particular musical arrangement. With the use of the mute switch a channel may be switched out of the mix situation without disturbing the settings of the individual mix level controls and once that microphone is required back in the system, then simply press the mute switch so that normal operation is once again accomplished.

**PHASE CONTROL**

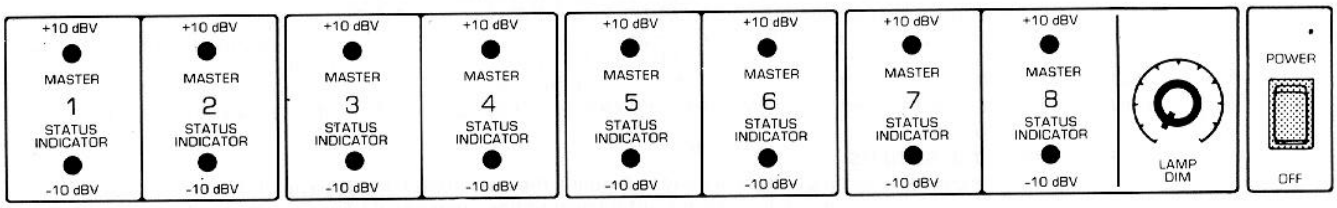
The phase switch is of the push/push type and simply "reverses the phase" of any signal being fed into an individual channel. Many times it is necessary to reverse the phase of microphones made by different manufacturers so that everything is once again in phase because there seems to be no standard and all microphones are not wired the same as they come from the different manufacturers. Feedback situations often dictate the need for phase reversal so that, in some cases, feedback may be reduced using phase reversal techniques.

**CHANNEL PFL CONTROL**

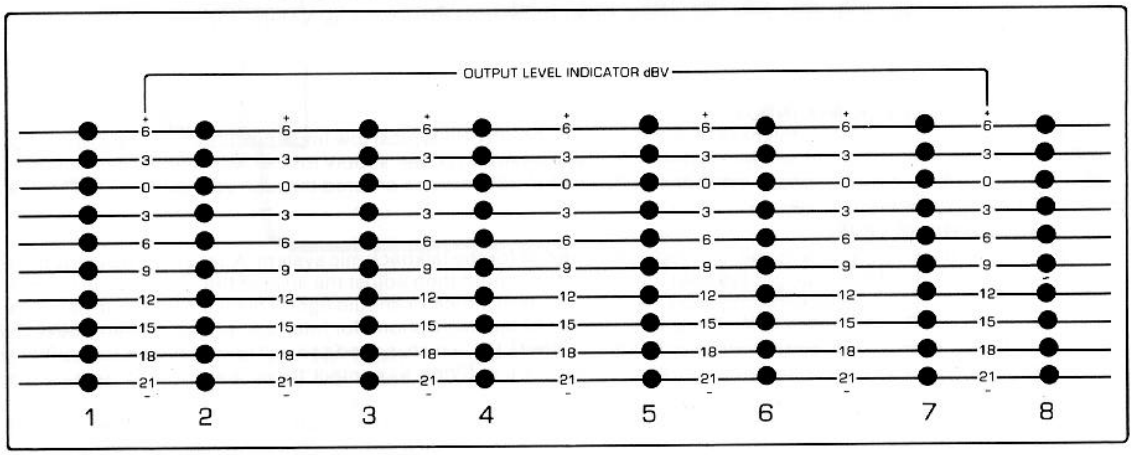
The PFL function allows the operator to instantly derive a "cue" (or prefade listen signal) through the headphones for any of the channels individually. Changes of equalization, input gain settings and any other channel functions may be monitored from the PFL system because the signal is derived after the equalization but before the channel matrix level controls. This function is also affected by the input gain setting. In order to monitor an individual channel for this function the PFL master level control in the Master section must be turned up to the desired listening level and PFL switch must be depressed on a channel. NOTE: ANY NUMBER OF CHANNELS MAY BE MONITORED AT THE SAME TIME WITH THE PFL SYSTEM.

**MATRIX LEVEL CONTROLS**

There are "eight" level controls on each channel of the Mark IV™ which complete the matrix and allow each instrument or vocal to be assigned to any or all of the **eight** mixes. The channel level controls should be operated near the "0" indication (or unity gain) whenever possible for optimum headroom and signal-to-noise. Operation of this control above the "0" point increases system gain (and noise) and below the "0" point sacrifices headroom. Notice each control is capable of 6 dB of gain above the ("0") unity gain position.




# PEAVEY MK-IV™ 16 ch. / 16 x 8 MONITOR MIXER




**SUMMING BUS STATUS INDICATION**

The individual mix systems have been provided with LED status indicators for each mix. The summing of several channels may cause an individual mix to become overloaded at the master summing amp. The controls provided to adjust the sum output of several channels are the individual level controls on each channel. With the LED indication appearing at -10 dBV and +10 dBV, it is a simple procedure to optimize the overall summed signal which is assigned, for instance to mix 1, and control the total output on those channels as they mix into the mix 1 summing system. Master mix outputs one through eight are identical as shown and once again the mix system is capable of +18 dBV and when the +10 LED flashes, it **doesn't mean that this particular mix has overloaded** but merely that +10 has been reached. (See description for Channel LED Status.)

**OUTPUT MIX LEVEL INDICATION**

Ten-segment LED arrays have been provided to visually indicate the output level of each of the Mark IV's™ separate mixes. Masters one through eight have their own individual LED ladders which indicate output levels from -21 dBV to +6 dBV. Each mix output is capable of +18 so there is **12 dB** of headroom **left** when the top LED flashes for each output section.

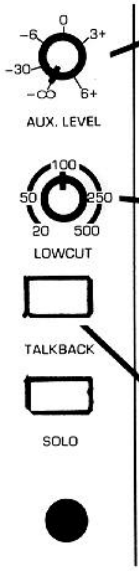
**LAMP DIM CONTROL**

This control allows the operator to adjust the lamp intensity at the mixer for each situation of house lighting. Clockwise rotation increases the intensity and counterclockwise dims lamps.

**AC POWER SWITCH/LED**

The power switch is a rocker-type which is off at the bottom position and on at the top. The LED indicates that the AC mains are supplying the necessary power to the mixer.

---




**MASTER MIX CONTROL FUNCTIONS**

**AUXILIARY LEVEL CONTROL**

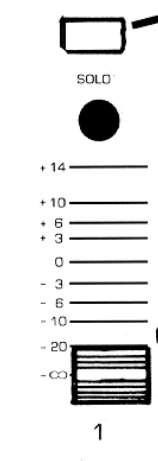
Each separate master mix is equipped with an auxiliary level control. The auxiliary level device controls the gain of each auxiliary input jack on the rear panel for each separate mix. For instance, when an additional mixer is added to the Mark IV™ Monitor and more channels are needed, the auxiliary unit may be patched into an auxiliary input for any one of the eight mixes and the auxiliary level control will regulate the gain of that system. The control is a rotary device with a "zero indication" for unity gain at the twelve o'clock position. Counterclockwise rotation of the auxiliary level will decrease sensitivity at the auxiliary input and clockwise rotation will increase the sensitivity with 6 dB of gain possible with this function.

**LOW CUT CONTROL**

Each of the eight separate masters includes a "low cut" control which tailors the low frequency response of all instruments or vocals assigned to each respective master mix. The purpose of this control is a "roll-off" function of 12 dB per octave beginning at the frequency which is selected by rotation of the control knob. The range of frequency selection is 20 Hz to 500 Hz. The low cut feature is a **very useful** device for matching the output signal with the capabilities of the power amp/speaker system for a particular mix. Low frequencies which the speaker system will not reproduce efficiently may be cut by 12 dB per octave thus eliminating the "muddiness" often encountered with monitor systems. Monitor speaker systems in general, are not required to reproduce as much low end as the "out-front" (main) house system. Many monitor speaker systems are incapable of reproducing frequencies much below 100 Hz and the durability of those systems is greatly increased when those devices **are not** driven with frequencies below their cut off capabilities.

**TALK-BACK FUNCTION**

The Talk-Back switch is a push/push type and allows talk-back assignment to any or all master mixes. This is a valuable feature which allows the operator to talk to any or all of the on-stage monitor systems which are in use. NOTE: IN ORDER TO ACTIVATE THE TALK-BACK SYSTEM, THE MOMENTARY TALKBACK ACTIVATE BUTTON IN THE TALK-BACK SECTION MUST BE DEPRESSED AND HELD DOWN. THIS FEATURE AVOIDS ACCIDENTAL OPERATION OF THE TALK-BACK SYSTEM.

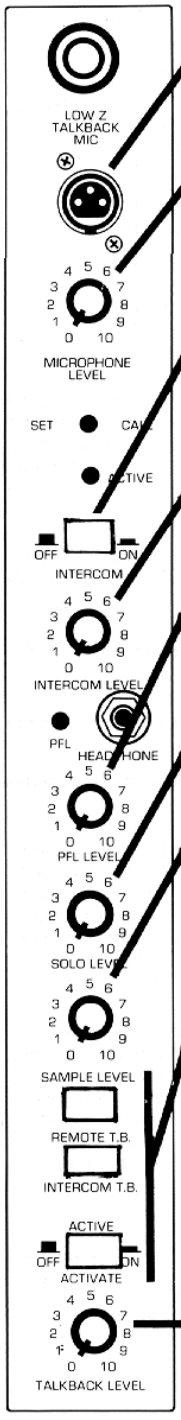



**SOLO FUNCTION**

The Solo switch is also a push/push type and allows the operator to select any or all master mixes in his headphone or sample output. Unlike the PFL function on each channel, the Solo function is the final signal output of each master mix after any master equalization and **after** the master slider controls. This allows the operator to listen to the actual mix quality and level going to the stage. In order to monitor any master with this function, the Solo Master Level control in the master section must be set to the desired listening level and a solo button depressed on any master mix. NOTE: ANY NUMBER OF MASTER MIXES MAY BE MONITORED AT THE SAME TIME WITH THE SOLO FUNCTION. ALSO THE PFL FUNCTION HAS PRIORITY OVER THE SOLO FUNCTION IN THAT IF A PARTICULAR PFL BUTTON IS DEPRESSED, ALL SOLO FUNCTION IS DEFEATED. THIS ALLOWS THE OPERATOR TO SELECT A PFL FUNCTION WITHOUT DEFEATING A SOLO FUNCTION.

**MASTER MIX LEVEL SLIDERS**

All eight master mix sliders function identically to regulate the level of each mix separately. For instance, Master 1 feeds the output on the rear panel for Mix 1 only. (See mix output description - rear panel.) These controls should be operated near the midpoint or "0" position for optimum headroom and signal-to-noise performance. NOTE: A SETTING ABOVE THE "0" dB POSITION WILL INCREASE OUTPUT SIGNAL BUT ALSO INCREASE SYSTEM NOISE AND A SETTING BELOW "0" dB WILL RESULT IN IMPROVED NOISE PERFORMANCE AT THE COST OF LESS HEADROOM. Notice there is 14 dB of gain available above the "0" position for situations where extra gain is required. A rough rule-of-thumb setting for the master sliders would be at the input sensitivity rating of the power amplifier for that particular mix. For instance, if the power amp sensitivity rating is +6 dBV, then adjust the slider for that mix to +6 dB as a starting point.




**LOW Z TALKBACK MIX INPUT**

This input is used for talkback and intercom. It should be a typical low impedance microphone of the operators choice. This input is unbalanced and incorporates a 15 volt phantom power supply for use with standard phantom powered mics if desired. (A special gooseneck microphone, Model EC-18™, may be ordered through your authorized Peavey dealer to work with this system.)

**MICROPHONE LEVEL**

The microphone level is the gain regulating device for the talkback mic system. After plugging in the microphone which is to be used for intercom and talkback, the operator should then adjust the level of the microphone until the green, active indicator is lit and under extreme levels the "set call" (red) indicator should light. This is the optimum gain adjustment for the microphone. The system is designed to indicate use of the microphone for intercom with the green, active LED and tapping on the microphone will usually result in lighting the red LED, which is used as a call function. This is to notify another person at the other end of the intercom system that "someone is trying to contact them" from the console.

**INTERCOM ON/OFF FUNCTION**

The push switch for intercom activation is a similar control to the PFL switch that merely assigns the talkback mic to the intercom system or removes it from the intercom function. This function allows the talkback features of the board to be separate from intercom. Intercom would normally be the communications between two mixing consoles, such as "stage monitors" and the "outfront board" or to the lighting personnel or to the show producer, etc. While talkback is usually the communication on stage from the monitor board to the players through the individual monitor mixes.

**INTERCOM LEVEL**

The intercom level is the gain regulating device which determines the listening level in the headphones for the intercom system and the related jack for the headphones is right below this function. The PFL LED which is located adjacent to the headphone jack provides visual indication when any PFL function switch is depressed. When all PFL switches are in the upward position, this PFL LED will not be lit.

**PFL LEVEL CONTROL**

PFL level control is a master function which sets the pre-fade listen level for any function that should be monitored individually through the headphones. This control should be adjusted for the desired listening level and the PFL push switch must be activated on a channel when pre-fade listen is desired. **The PFL level control overrides the solo level function** and anytime a PFL push switch is activated this function is monitored in both sides of a typical stereo headphone set up. (NOTE: Intercom overrides both PFL and solo system.)

**SOLO LEVEL CONTROL**

The solo level control is the gain determining element for the entire solo system. When a solo push switch is depressed on any of the eight mixes, the solo level controls the amount of signal feeding the headset. The solo system allows the mix engineer to monitor only one mix or all eight at the same time, according to how many solo switches are pressed.

**SAMPLE LEVEL**

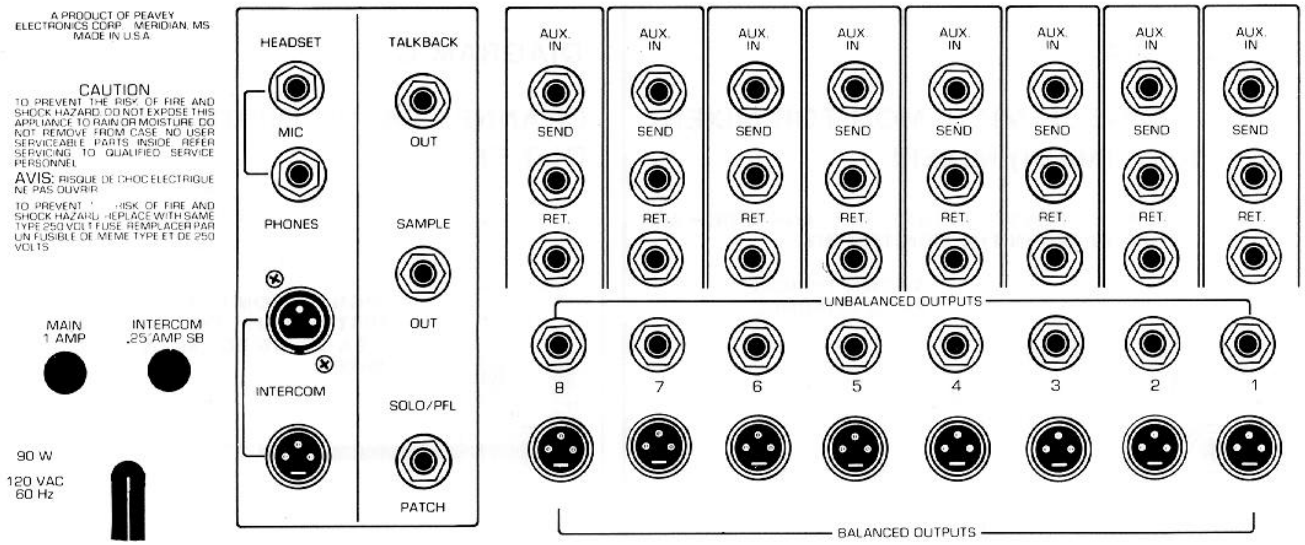
Many times monitor applications require the monitor mix engineer to have a sampling amp/speaker system in order to listen to the exact same mix that the players are hearing. The sample level control is the gain determining element for the sample output on the rear panel. Some monitor situations may only call for headphone sampling at the monitor mix board and in that case the sample output would not necessarily be used. In most cases, however, for high level operation the separate sampling speaker is much more effective than typical headphones. The main advantage is that the monitor mix engineer is able to listen under similar conditions through the same type speaker as the performers on stage. NOTE: BOTH SOLO LEVEL AND PFL LEVEL CONTROLS MUST BE PRESET BEFORE SAMPLE OUTPUT CAN OCCUR.

**TALKBACK SECTION**

There are three "push-type" switches associated with the talkback section. Two of those switches remain **down** when pressed until pressed a second time and the talkback off/on switch is a "momentary switch" that must be held down when speaking on the talkback system to prevent accidental operation. The **remote talkback** push switch allows signal to be sent to the remote talkback output on the rear panel. This feature may be used for a separate talkback system in an adjacent room or other location in the building but requires a power amplifier and a speaker system of its own. When communication, announcements, etc. are desired through the remote facility the system must be patched through the remote talkback output and the **talkback level** regulates the gain of that function. This output may also be used for patching to another monitor mixer (when additional channels are required) and allows talkback into that mixer. The **intercom talkback** push switch allows the talkback system which functions through all eight separate mixes to be interfaced with an outfront (main) console. For instance, the outfront engineer during sound checks, etc. needs to communicate with the musicians on stage. When this situation arises, the intercom talkback push switch should be depressed thus locking the talkback system into the intercom function of the main house board. Then simply by speaking in the normal procedure on the intercom system, the house mix engineer may communicate through any or all monitor mixes on stage according to which talkback switches have been depressed for the different mixes. There is an LED that indicates when the talk system is active and it is located directly above the talkback off/on momentary switch.

**TALKBACK LEVEL**

Talkback **level** regulates the gain of the talkback system. Clockwise rotation of this control allows the mix engineer to increase the level of the talkback signal and counterclockwise rotation decreases the level of the talkback function.




## REAR PANEL

### AUXILIARY INPUT JACK

There is an auxiliary input for each separate mix to allow patching in other "out board" signals such as more monitor channels for a particular mix, another monitor console, etc. (See description of Auxiliary Level Control.)

### SEND AND RETURN JACKS

Each separate mix includes a send and return jack for in-line patching of equalizers, compressors, limiters, etc. This unique patching capability allows outboard equalization to be interfaced with the monitor console **before the solo function**. Monitor engineers usually prefer to listen to their sampling speaker system after the outboard equalization and this send and return feature provides just that. (Note: The return jack is of the switching type.) Note: This patch is before (pre) the summing bus status indication.

### MIX OUTPUTS

Each separate mix has a 600 ohm transformer balanced output (3 pin XLR) and a 600 ohm unbalanced (1/4" phone jack) output for patching out to the various power amp/speaker systems.

## CHANNEL INPUTS

### LOW Z BALANCED IN/OUT JACKS

Each channel is equipped with a low Z (600 ohm transformer balanced) splitter system for the interconnection of microphones or instruments between the house (main) mixer and the Mark IV™ Monitor console. The splitter system allows mics, which need to be heard in the monitor mix, to be patched directly to the low Z (in) jack and the house mixer receives the signal from those same microphones from the low Z (out) jack through the main snake cable system. (See Diagram A.)

### CHANNEL PATCH JACK

The channel patch is yet another way into each channel (unbalanced) after the transformer. This jack allows unbalanced line levels to be connected directly to the monitor console and those signals may then be taken transformer balanced from the low Z out jack to the house (main) mixer. The channel patch may also be used as a low to high impedance conversion when a channel is not being used for the monitor system (See Diagram C).

### PRE SEND AND RETURN JACKS

The pre send and return jacks allow individual channel patching immediately **after** the input gain control and **before** the equalization. The send jack may be used to patch out from the channel for any reason a separate output is needed and the audio chain is not interrupted. The return jack is of the switching type and breaks the normal path of signal flow thus allowing limiters, equalizers, noise gates, etc. to be patched from the pre out back into the pre return jack, for individual channel conditioning.

### HEADSET (MIC/PHONE) JACKS

These jacks are 1/4" phone and are designed to accept typical communications headsets that are equipped with mic and headphones. The com-system is monaural, and should accept most commercially available intercom mics and headphones. (These units may be used in place of a separate talkback mic.)

### INTERCOM PATCH JACKS

Two 3-pin XLR type jacks have been provided for connecting intercom system between mixers or "Talk-Com™" units. Master-Com™ and Talk-Com™ units are soon to be available from Peavey Electronics. The intercom system which is built into the Mark IV™ Monitor is equivalent to what will be later known as the "Master Com™" available as a separate unit. (See Diagram E.)

### TALKBACK OUTPUT

The talkback output jack is 1/4" phone and is designed to be used when additional talkback capability is required other than to the stage monitor mix situation. There are many uses for additional talkback capability around sound reinforcement situations, such as a separate paging system with amplifier and speaker in the dressing room to notify groups that it is showtime etc. This output can also be patched to an auxiliary input of a second mixer for talkback capability to that mixer if desired.

### SAMPLE OUTPUT JACK

The sample output jack is 1/4" phone and provides a signal from the sample level on the front panel and should be used for monitor mix engineer to sample the different mixes that are happening on stage. This creates a separate system which is usually located near the monitor mix console and should include the same basic type of amplifier and speaker system that is employed on stage so that the mix engineer will know exactly how the monitors sound on stage to the performing musicians. The patch sequence for this system would be from the sample output jack to the amp/speaker system that is used for the sampling source. NOTE: FOR MOST HIGH LEVEL MONITOR APPLICATIONS A SEPARATE SAMPLING SPEAKER/AMPLIFIER SYSTEM IS MUCH MORE EFFECTIVE THAN TRYING TO DO A MONITOR MIX THROUGH HEADPHONES. (See explanation of Sample Level Control.) (See Diagram D.)

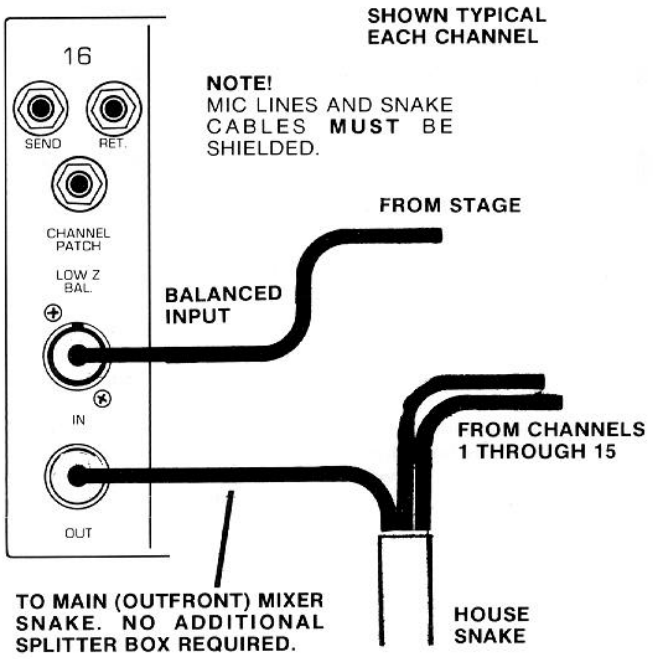
### SOLO/PFL PATCH

The solo/PFL patch is stereo 1/4" phone and is designed to interconnect the logic of the solo feature and the PFL feature between two consoles such as two Mark IV™ Monitors. When two consoles are used for additional channels in a monitor situation, the **logic** should be connected so that listening capability is possible to both consoles through the PFL function of the master console. Also the solo function must be interconnected so that from the master console you will be able to listen to individual mixes of the auxiliary console. A stereo 1/4" phone plug is required to make these connections at the solo/PFL patch.

## DIAGRAM A

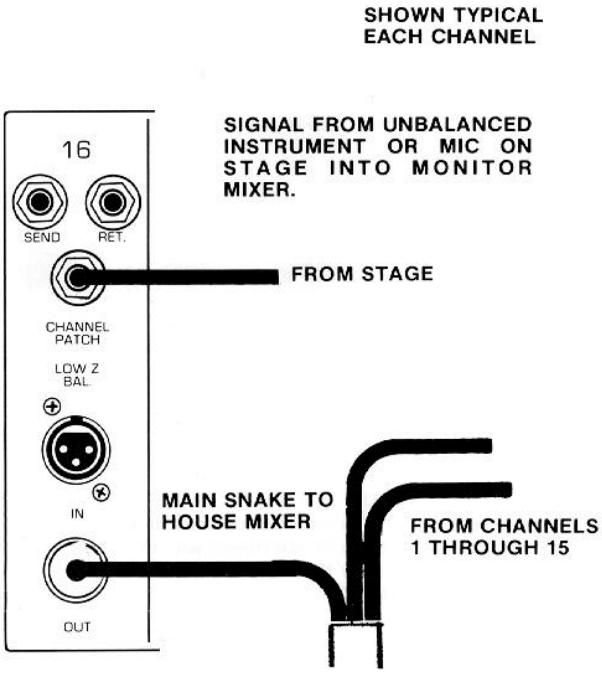
## PATCH BETWEEN MICS, MONITOR MIXER AND HOUSE (MAIN) MIXER!

SIGNALS FROM MICS OR INSTRUMENTS ON STAGE WHICH ARE REQUIRED IN THE MONITOR MIX



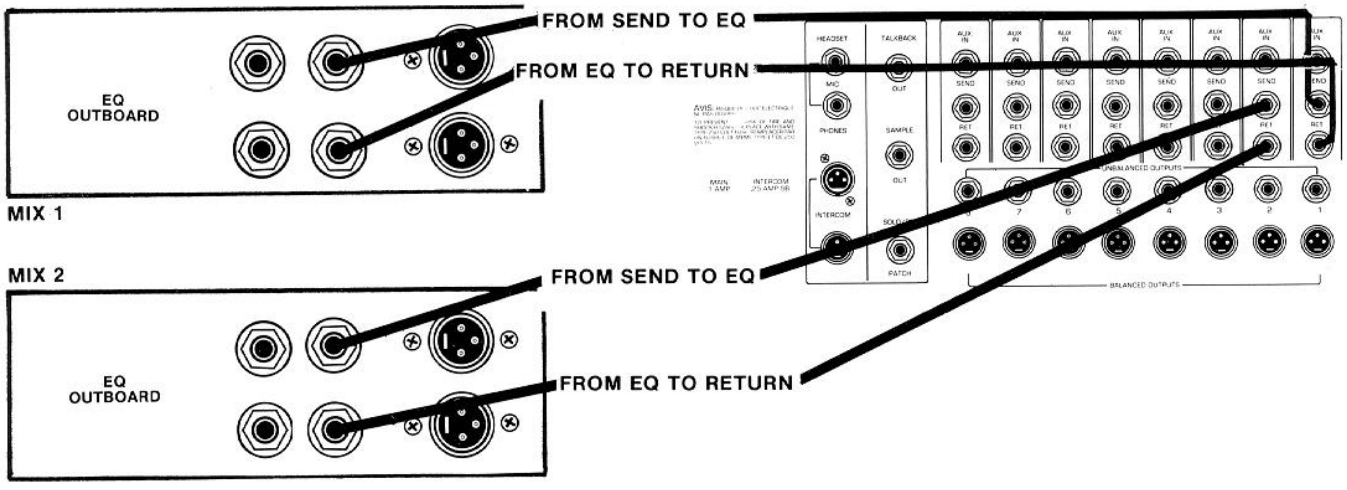

## DIAGRAM C

## CHANNEL PATCH INPUT/BALANCED OUT FEATURE




## DIAGRAM B

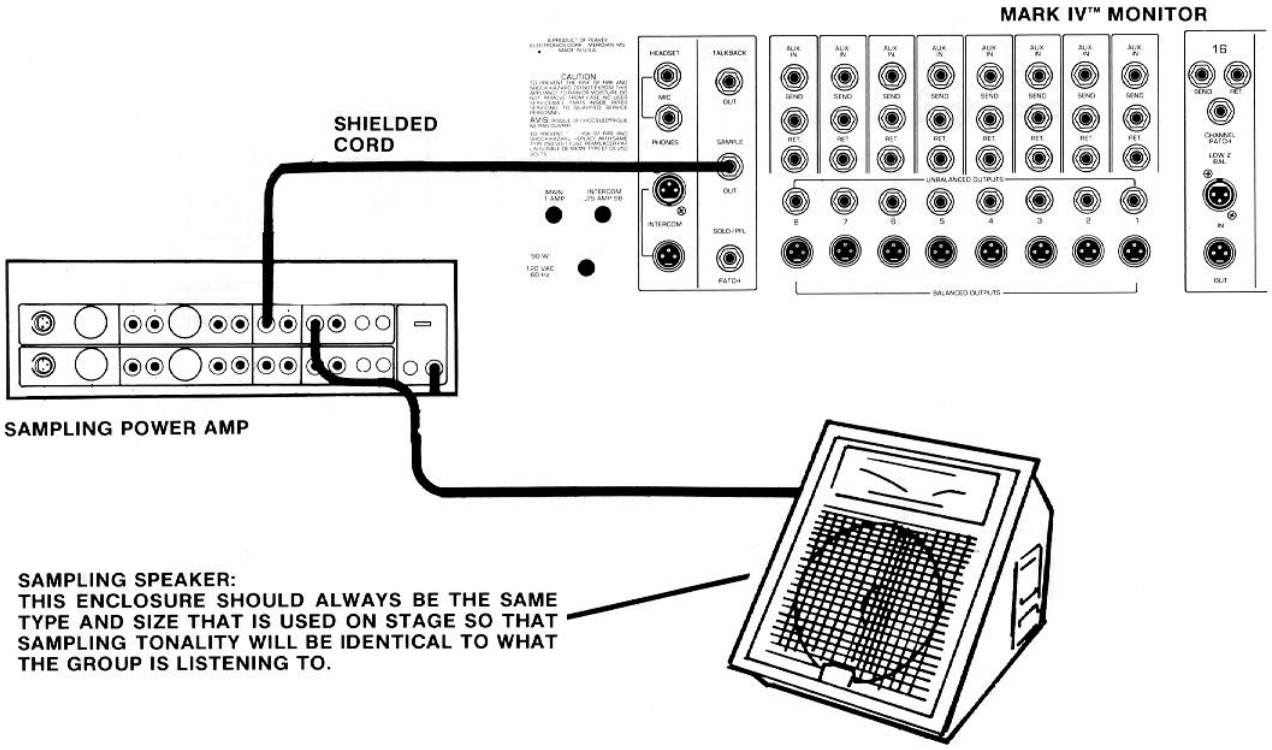
## EXTERNALLY EQUALIZED MONITOR MIXES



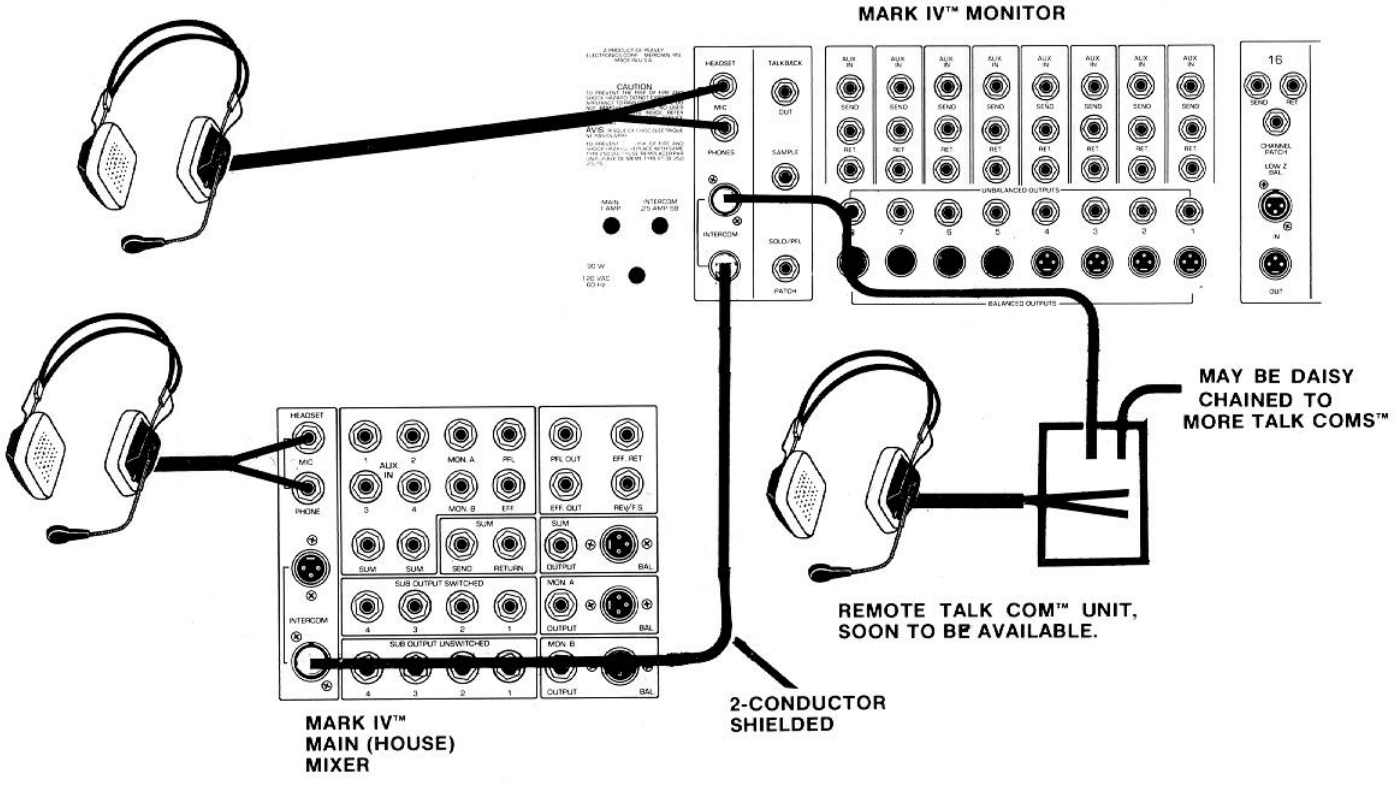

**NOTES!**

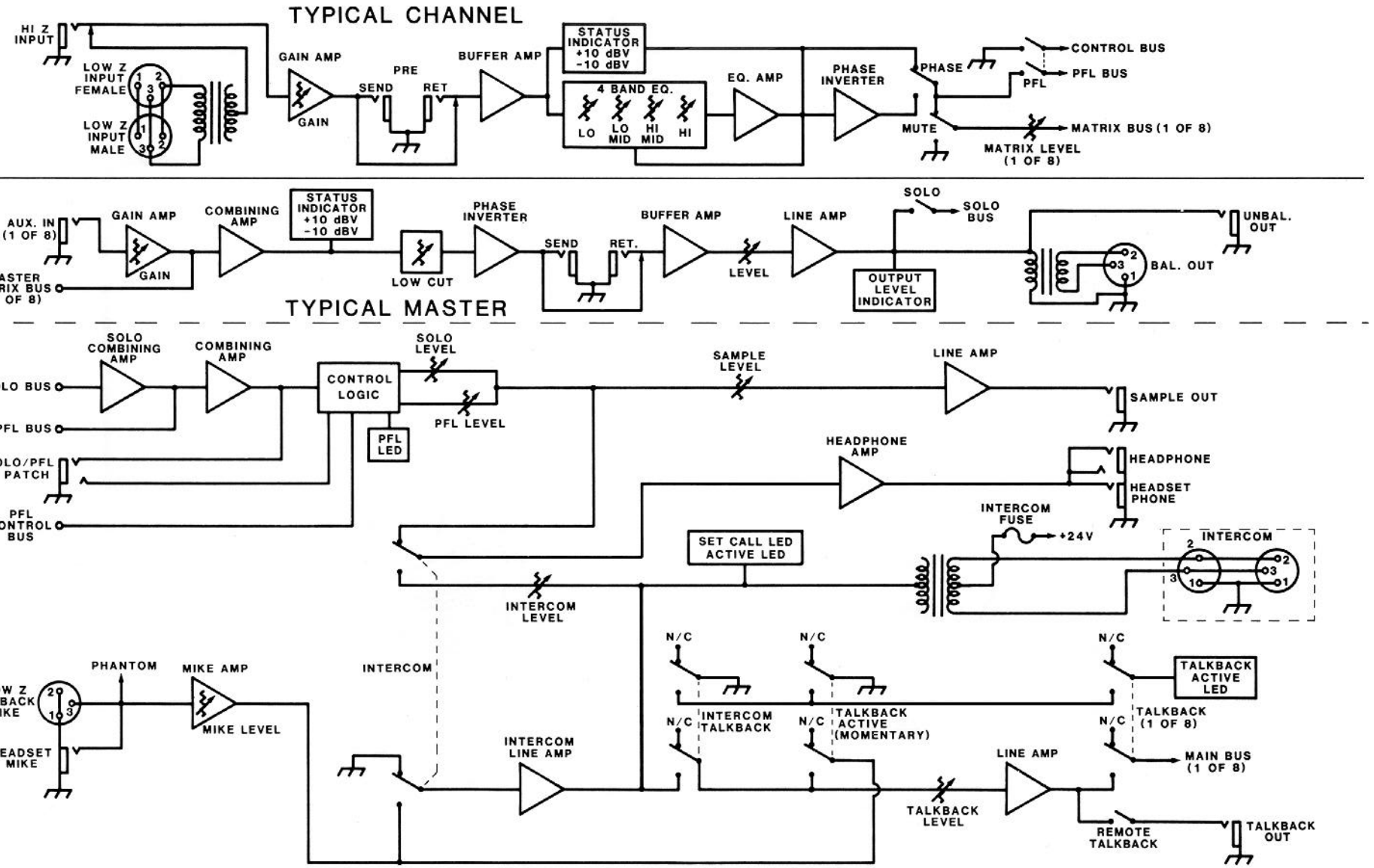
- Use good quality shielded cords fpr all connections.
- Only (2) equalizers are shown, but system is capable of (8) equalized mixes.
- By patching outboard EQ into send and return jacks, the engineer may listen to the equalized signals with Solo/Sampling system.
- Use 1/3 octave equalizers whenever possible.

## DIAGRAM D SAMPLE SPEAKER/POWER AMP PATCH




## DIAGRAM E INTERCOM PATCH




TYPICAL CHANNEL
HI Z INPUT
LOW Z INPUT FEMALE
LOW Z INPUT MALE
GAIN AMP
GAIN
PRE
SEND
RET
BUFFER AMP
STATUS INDICATOR +10 dBV -10 dBV
4 BAND EQ.
LO
LO MID
HI MID
HI
EQ. AMP
PHASE INVERTER
PHASE
CONTROL BUS
PFL
PFL BUS
MUTE
MATRIX BUS (1 OF 8)
MATRIX LEVEL (1 OF 8)
AUX. IN (1 OF 8)
GAIN AMP
GAIN
COMBINING AMP
STATUS INDICATOR +10 dBV -10 dBV
LOW CUT
PHASE INVERTER
SEND
RET.
BUFFER AMP
LEVEL
LINE AMP
SOLO
SOLO BUS
OUTPUT LEVEL INDICATOR
UNBAL. OUT
BAL. OUT
MASTER MATRIX BUS (1 OF 8)
TYPICAL MASTER
SOLO COMBINING AMP
COMBINING AMP
SOLO BUS
PFL BUS
SOLO/PFL PATCH
PFL CONTROL BUS
CONTROL LOGIC
SOLO LEVEL
PFL LED
PFL LEVEL
SAMPLE LEVEL
LINE AMP
SAMPLE OUT
HEADPHONE AMP
HEADPHONE
HEADSET PHONE
INTERCOM FUSE
+24V
INTERCOM
SET CALL LED ACTIVE LED
INTERCOM LEVEL
N/C
TALKBACK ACTIVE LED
INTERCOM TALKBACK
TALKBACK ACTIVE (MOMENTARY)
TALKBACK (1 OF 8)
MAIN BUS (1 OF 8)
PHANTOM
MIKE AMP
LOW Z TALKBACK MIKE
HEADSET MIKE
MIKE LEVEL
INTERCOM
INTERCOM LINE AMP
LINE AMP
TALKBACK LEVEL
REMOTE TALKBACK
TALKBACK OUT

# Specifications:

**SUMMARY OF FUNCTIONS:**
16 channels in; 8 mixes out; 1 sample out; bridging male and female XLR transformer balanced inputs with ground isolation, Hi Z patch input, and pre send & return each channel: 1 transformer balanced & 1 unbalanced with EQ send & return each output; intercom system with phantom powered slaved; mono headphone out; phantom power gooseneck mic input; communication type headset option; talk-back to all mixes; intercom to talk-back capability; input gain/attention, 4-band EQ, mute, PFL, & phase reverb each channel with LED status indicator; solo, variable Lo cut, master slider, LED ladder, auxiliary input and mix LED status indicators each out

**INPUTS, EACH CHANNEL:**
Bridging male & female Low Z transformer balanced microphone with ground isolation; 1 High Z unbalanced patch; 1 High Z unbalanced pre return

**INPUTS, MASTER:**
1 High Z unbalanced auxiliary, 1 High Z unbalanced effects return for each of 8 mixes; 1 unbalanced PFL patch

**OUTPUTS, EACH CHANNEL:**
1 Low Z unbalanced pre send line

**OUTPUTS, MASTER:**
1 Low Z balanced line, 1 Low Z unbalanced line &
1 Low Z unbalanced effects send for each of 8 mixes,
1 Low Z unbalanced sample out; 1 Low Z unbalanced talk out,
1 Low Z headphone out;
2 (1 male/1 female) balanced intercom lines

**CHANNEL MICROPHONE INPUTS:**
Mic Impedance: Low Z 600 ohms transformer balanced
Nominal Input Level: -22 dBM, 60 mV RMS
Minimum Input Level: -38 dBM, 10 mV RMS
Maximum Input Level: +22 dBM, 10 volts RMS

**CHANNEL PATCH (HIGH Z MIC) INPUTS:**
Line Impedance: Hi Z 50K ohms unbalanced
Nominal Input Level: -14 dBV, 200 mV RMS
Minimum Input Level: -30 dBV, 30 mV RMS
Maximum Input Level: +30 dBV, 30 volts RMS

**CHANNEL PRE RETURN AND MASTER EFFECT RETURN INPUTS:**
Line Impedance: Hi Z 33K ohms unbalanced
Designed Input Level: 0 dBV, 1 volt RMS

**MASTER AUXILIARY INPUTS:**
Line Impedance: Hi Z 10K ohms unbalanced
Nominal Input Level: 0 dBV, 1 V RMS
Minimum Input Level:-6 dBV, 0.5 V RMS
Maximum Input Level: +30 dBV, 30 V RMS

**MASTER BALANCED OUTPUTS:**
Load Impedance: 600 ohms or greater
Nominal Output: - dBV, 1 V RMS
Maximum Output: +18 dBV, 8 V RMS into 50K ohms load
Maximum Output: +16 dBM, 5 V RMS into 600 ohms load

**MASTER, CHANNEL SENDS, MASTER EFFECT SENDS, & SAMPLE UNBALANCED OUTPUTS:**
Load Impedance: 600 ohms or greater
Nominal Output: 0 dBV, 1 V RMS
Maximum Output: +18 dBV, 8 V RMS into 50K ohms load
Maximum Output: +14 dBM, 4 V RMS into 600 ohms load

**HEADSET: (TALK-BACK & COMMUNICATION/PFL FUNCTIONS)**
Compatible with 150 ohm mic, 600 ohm receiver with 1/4" phone plugs for each

**HEADPHONES: (MONO AMP WITH STEREO JACK)**
Load Impedance: 4 ohms to 50 ohms
Maximum Output Power: 100 mW

**TALK-BACK MIC INPUT:**
XLR connector with +15 V phantom power compatible with electret gooseneck type mic

**INTERCOM LINE:**
Bridging, balanced XLR's with 24 V phantom power for interface with external slave Talk-Com™ and other mixing consoles with compatible features

**INTERCOM SPECS:**
Nominal Impedance: 600 ohms transformer balanced
Nominal Level: 0 dBV, 1 V RMS

**THE FOLLOWING SPECS MEASURED WITH A NOMINAL INPUT GAIN SETTING OF +20 dB ALL CHANNELS, ALL CHANNEL MATRIX CONTROLS AND ALL MASTER SLIDER CONTROLS ARE SET AT 0 dB, ALL EQ SET FLAT, ALL LO Z INPUTS TERMINATED WITH 600 OHMS, ALL HI Z INPUTS & ALL OUTPUTS TERMINATED WITH 47K OHMS.**

**FREQUENCY RESPONSE: (Any In/Out Combination with 1 V RMS Output)**
+0, -2 dB, 20 Hz to 30 KHz

**SYSTEM HUM & NOISE: (All Channels On)**
-84 dBV, (Hi Z line inputs)
-80 dBV, (Lo Z mic inputs)

**EQUIVALENT INPUT NOISE:**
**(20 Hz - 20 KHz, 150 ohms)**
**-127 dBV**

**OVERALL DISTORTION: (Any In/Out Combination, 20 Hz - 20 KHz @ 1 V RMS)**
Less than .05% THD, typically below .01%

**EQUALIZATION:**
+-15 dB @ 60 Hz & 15 KHz, shelving
+-15 dB @ 400 Hz & 3 KHz, peak/notch

**CROSSTALK:**
Greater than 70 dB @ 1 KHz

**MAXIMUM AVAILABLE GAIN:**
+14 dB Master Slider
+6 dB Channel Matrix Control
+30 dB Input Control
+17 dB Balanced Input Transformer
+67 dB Total

**LED READOUT RANGE: (For Each of 8 Mixes)**
-21 to +6 dBV

**CHANNEL & SUB STATUS INDICATOR LED CALIBRATION:**
-10 dBV (Green) & +10 dBV (Red)

**POWER REQUIREMENTS:**
120 VAC, 60 Hz, 90 watts

Due to our efforts for constant improvement, features and specifications listed herein are subject to change without notice.

## MIXER CAUTIONS

### DANGER

EXPOSURE TO EXTREMELY HIGH NOISE LEVELS MAY CAUSE A PERMANENT HEARING LOSS. INDIVIDUALS VARY CONSIDERABLY IN SUSCEPTIBILITY TO NOISE INDUCED HEARING LOSS, BUT NEARLY EVERYONE WILL LOSE SOME HEARING **IF** EXPOSED TO SUFFICIENTLY INTENSE NOISE FOR A SUFFICIENT TIME.

THE U.S. GOVERNMENT'S OCCUPATIONAL SAFETY AND HEALTH ADMINISTRATION (OSHA) HAS SPECIFIED THE FOLLOWING PERMISSIBLE NOISE LEVEL EXPOSURES:

| DURATION PER DAY IN HOURS | SOUND LEVEL dBA, SLOW RESPONSE |
|---|---|
| 8 | 90 |
| 6 | 92 |
| 4 | 95 |
| 3 | 97 |
| 2 | 100 |
| 1½ | 102 |
| 1 | 105 |
| ½ | 110 |
| ¼ or less | 115 |

ACCORDING TO OSHA, ANY EXPOSURE IN EXCESS OF THE ABOVE PERMISSIBLE LIMITS COULD RESULT IN SOME HEARING LOSS.

EAR PLUGS OR PROTECTORS IN THE EAR CANALS OR OVER THE EARS MUST BE WORN WHEN OPERATING THIS AMPLIFICATION SYSTEM IN ORDER TO PREVENT A PERMANENT HEARING LOSS **IF** EXPOSURE IS IN EXCESS OF THE LIMITS AS SET FORTH ABOVE. TO INSURE AGAINST POTENTIALLY DANGEROUS EXPOSURE TO HIGH SOUND PRESSURE LEVELS, IT IS RECOMMENDED THAT ALL PERSONS EXPOSED TO EQUIPMENT CAPABLE OF PRODUCING HIGH SOUND PRESSURE LEVELS SUCH AS THIS AMPLIFICATION SYSTEM BE PROTECTED BY HEARING PROTECTORS WHILE THIS UNIT IS IN OPERATION.

### CAUTION

THIS MIXING CONSOLE/PREAMP HAS BEEN DESIGNED AND CONSTRUCTED TO PROVIDE ADEQUATE SIGNAL (VOLTAGE) FOR PLAYING MODERN MUSIC. IMPROPER USE OF THE GAIN/EQUALIZATION CONTROLS AND/OR IMPROPER USE OF INTERNAL/EXTERNAL BUSSES **MAY** CREATE CLIPPING (SQUARE WAVES) AND POSSIBLY CAUSE SUBSEQUENT DAMAGE TO THE LOUDSPEAKER SYSTEMS. EXTENDED OPERATION OF THE GAIN/EQUALIZATION CONTROLS IN THEIR MAXIMUM POSITIONS IS, THEREFORE, **NOT** RECOMMENDED. PLEASE BE AWARE THAT **MAXIMUM POWER** CAN BE OBTAINED WITH VERY LOW SETTINGS OF THE **GAIN/EQUALIZATION** CONTROLS IF THE INPUT SIGNAL IS VERY STRONG.

**PEAVEY ELECTRONICS CORPORATION**
711 A Street / Meridian, MS 39301 / U.S.A.
© 1983 Printed in U.S.A.

80370396 2/83